USER CONTROLLED AUDIO QUALITY FOR
VOICE-OVER-IP TELEPHONY SYSTEMS

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BACKGROUND

Public Switched Telephone Network (PSTN) telephony offers only one level of

service based on the use of a single audio quality regime. For example, the PSTN uses 64

kilobits Per Second (BPS) Pulse Code Modulated (PCM) audio with a 300-3400 Hertz

15   passband. PSTN telephony further offers only one control regime that deterministically

either grants or denies service rather than degrading service when a failure or network

congestion occurs.

Voice-Over-Internet Protocol (VoIP) systems are often built to mimic this service

regime but are also inherently more flexible. Certain VoIP protocols and algorithms can

20   adapt to varying service levels. Example adaptations include either denying service or

degrading service when resources cannot be reserved, or reserved resources become

unavailable. Different encoder/decoders (Codecs) are switched in and out if bandwidth

becomes more or less scarce. Forward Error Correction (FEC) can be added or removed as

packet error rates change. Packetization intervals can also be varied to either limit bandwidth

25   utilization or limit packet transmission rate.

To date these adaptations have been statically provisioned by the network designer

and are not user controllable. The problem is that these adaptations may not be optimized for

the current telephone users or the current VoIP call. For example, a given quality of service

may be more than adequate for one user. However, that same given quality of service may be

30   unsatisfactory to a different user.

5   The present invention addresses this and other problems associated with the prior art.

SUMMARY OF THE INVENTION

A call adaptation system tracks adaptation schemes used for transmitting audio

packets in a Voice Over IP call. A user response to the Voice Over IP (VoIP) call is

monitored. The call adaptation system then dynamically varies the adaptation schemes used

10   for transmitting the audio packets according to the monitored user response.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a diagram of a communications network including a call adaptation system.

FIG. 2 is a detailed diagram of the call adaptation system shown in FIG. 1.

15   FIG. 3 is a block diagram showing how the call adaptation system modifies different

adaptation parameters.

FIG. 4 is a block diagram showing how the call adaptation system modifies different

adaptation parameters according to network congestion measurements.

FIG. 5 a block diagram showing how the call adaptation system operates in a Plain

20   Old Telephone System.

FIG. 6 a block diagram showing the call adaptation system is implemented with a

graphical user interface.

DETAILED DESCRIPTION

25   . FIG. 1 shows a communications network 12 that includes a packet switched network

26. The packet switched network 26 includes multiple network processing devices 28 that

connect Voice Over Internet Protocol (VoIP) calls between different telephony end points.

For example, a VoIP call 22 is established between endpoint 14 and endpoint 30. The

5   network processing devices 28 can be any routers, switches, gateways, concentrators, etc.,

used for transferring packets over the packet switched network 26.

A call adaptation system 16 includes a user input device 18 that receives inputs

directly from a user 32. The call adaptation system 16 allows the user 32 to dynamically

control audio quality by dynamically varying call adaptation parameters used for conducting

10   the VoIP call 22.

The user 32 initiates the VoIP call over the telephone endpoint 14. The user 32 then

communicates with endpoint 30 through handset 34. If there is a desire to vary the perceived

audio quality of the VoIP call 22, the user 32 adjusts input device 18. For example, the user

32 may perceive the VoIP call 22 as having unsatisfactory sound quality. The user can turn

15   input device 18 in a counter-clockwise direction in an attempt to increase sound quality of the

VoIP call 22. The call adaptation system 16 then varies call adaptation parameters that

improve sound quality.

In another example, the user 32 may think the sound quality of the VoIP call 22 is just

fine. However, the user 32 wants to reduce the cost of the VoIP call 22. The user 32 turns

20   the input device 18 in a clockwise direction in an attempt to reduce call cost. The call

adaptation system 16 then modifies the call adaptation parameters, such as used bandwidth,

to reduce the cost of the VoIP call 22. These adjustments to the adaptation parameters may

reduce the sound quality of the VoIP call 22.

The user 32 can control how hard the call adaptation system 16 tries to vary the

25   current sound quality in the VoIP call 22. The user 32 can control the adaptation parameters

both before and during the VoIP call 22.

The input device 18 differs depending on the instrument used for establishing the

VoIP call 22. For example, a physical knob or screen control may be used on an IP phone. A

5   graphical slider or similar icon representation may be used on a Personal Computer (PC)

phone. The input device 18 can have two extreme settings and any number of intermediate

settings. Examples of some of these embodiments are shown below in FIGS. 5 and 6.

FIG. 2 is a detailed diagram of the telephony endpoint 14 and the call adaptation

system 16 shown in FIG. 1. A phone 36 generates audio signals 44 that are encoded into

10   digital data by a voice encoder 38. A packetizer 40 converts the digital data into Voice over

IP (VoIP) packets that are then transmitted over the packet switched network 26 through a

transmitter interface 42.

The voice encoder 38, packetizer 40 and transmitter 42 use adaptation parameters 44

for encoding and packetizing the audio signals from phone 36. For example, different voice

15   coder algorithms 46 are used by voice encoder 38 to convert the analog audio signals 44 into

digital data. Two examples of voice coders are International Telecommunications Union

(ITU) Standards G.723.1 and G.711. These different voice coders 46 encode different

amounts of audio signals into digital data and ultimately vary the sound quality of the VoIP

call 22.

20   In another example, different packet payload sizes 48 can be used by packetizer 40

when converting the digital data from voice coder 38 into VoIP packets. These packet

payload sizes 48 can vary the amount of delay experienced during the VoIP call 22. Varying

packet payload according to measured network congestion is described in copending U.S.

Patent Application, Ser. No. 09/181,947, filed October 28, 1998, entitled: CODEC-

25   INDEPENDENT TECHNIQUE FOR MODULATING BANDWIDTH IN PACKET

NETWORK.

Another adaptation parameter is the type of Forward Error Correction (FEC) 50 used,

if any, to prevent packet loss during the VOIP call 22. A bandwidth reservation parameter 52

5   identifies an amount of bandwidth, if any, the endpoint 20 attempts to reserve on the packet

switched network 26 for the VoIP call 22.

A call adaptation controller 54 in the call adaptation system 16 adjusts the adaptation

parameters 44 according to the user response detected from input device 18. The call

adaptation controller 54 can also vary the adaptation parameters 44 according to the

10   congestion measurements 24 received from packet switched network 26. This is described

below in FIG. 4.

It should be understood that this is just an example of the types of adaptation

parameters 44 that may be used to adjust the sound quality of the VoIP call 22. The call

adaptation system 16 can use other parameters that vary the sound quality of the VoIP call

15   22.

Mid-Call Resource Reservation

FIG. 3 shows in further detail how the call adaptation system 16 isolates the user 32

from network congestion by initiating or terminating resource reservation and/or by

20   modulating packetization, forward error correction, and the codec. The call adaptation

system 16 decides mid-call whether to reserve network resources for the VoIP call or to use a

best effort scheme according to the monitored user response. For example, at a low sound

quality user setting, the VoIP call may use a best effort scheme for establishing and

conducting the VoIP call. At a higher sound quality user setting, the VoIP call may try to

25   reserve network resources.

User response to the VoIP call is continuously monitored in block 56. If a low sound

quality request is detected in block 58, the call adaptation controller 54 (FIG. 2) uses best

effort packet traffic to establish or continue the current VoIP call. Best effort packet traffic is

5   defined as VoIP packets that are transmitted with no pre-reservation of network resources.

Thus, if a network processing device in the packet based network is congested, some or all of

the VoIP packets may be dropped before reaching their destination point.

With the low sound quality request, a low quality voice coder is selected by the call

adaptation system. The low quality voice coder uses a relatively small number of digital bits

10   to represent a given portion of an audio signal. At the low sound quality setting no Forward

Error Correction (FEC) and relatively long packet payloads might be used in the VoIP call.

If the user makes a first intermediate sound quality request in block 60, the adaptation

system initiates a request to reserve network resources but also accepts best effort traffic

while trying to reserve the network resources. One protocol used for reserving network

15   resources is the Resource Reservation Protocol (RSVP). The RSVP protocol works through a

protocol exchange which visits each network processing node along the VoIP media path.

The RSVP messages request network processing devices along a media path to reserve a

particular packet bandwidth.

20   The call adaptation system provides a unique resource reservation scheme that

launches resource reservation mid-call. The first intermediate voice quality request detected

in block 60 launches the RSVP reservation request mid-call in block 64. If the RSVP

reservation request is accepted by all the network processing nodes in the media path, then

the call adaptation system has locked down the resources for providing higher voice quality.

25   If the user then requests even higher sound quality, the call adaptation system can increase

the packet rate and be assured the network can handle the increased bandwidth requirement

for the VoIP call.

If the RSVP request is not accepted, then the call adaptation system may not be able to improve voice quality if the user requests improved sound quality. In this case, there may be no improvement in sound quality if the user increases the voice quality request. The call adaptation system in block 64 may also switch to a better quality voice coder and/or reduce the packet payload size.

If a second intermediate sound quality request is detected in block 66, best effort traffic may still be used but a better quality voice coder and/or even shorter packet payloads may be used in block 68. In combination with the other adaptation parameter modifications, or by itself, FEC may be added to the VoIP call.

If a high sound quality request is detected in block 70, the VoIP call is conducted using only reserved network resources in block 72. A highest quality voice coder is used along with possibly an even shorter packet payload and FEC. The call adaptation system then continues to monitor for new user sound quality requests in block 56.

The example described above is for illustrative purposes. As described above, alternative implementations can also be used. For example, each user setting may include modification of a different one or different combinations of the adaptation parameters described above. Each setting may modify just one or a subset of the adaptation parameters described above. Alternatively, additional settings may provide smaller changes to the adaptation parameters or charge adaptation parameters not mentioned above.

Modifying Adaptation Parameters Based On Congestion Measurements

Referring briefly back to FIG. 1, to improve voice quality, it may be tempting to simply "crank up the gain" by injecting a higher packet rate into the VoIP call 22. However, this can actually increase congestion in the packet network 26 and be counter-productive to

5   improving audio quality. The call adaptation system 16 takes congestion measurements 24

into account through jitter and/or packet loss measurements. These congestion measurements

24 are conveyed back to the call adaptation system 16 using existing control protocols such

as the Real Time Control Protocol (RTCP). The call adaptation system 16 takes the network

congestion measurements 24 into account when adjusting the adaptation parameters to

10   improve responsiveness to the user sound quality requests.

FIG. 4 shows how the call adaptation system 16 uses the congestion measurements 24

to determine what adaptation parameters 36 are used for varying the quality of the VoIP call

22. Referring to both FIG. 1 and FIG. 4, congestion measurements 22 are monitored in block

61 and the user response to the current VoIP call is monitored in block 63. In one example,

15   the call adaptation system 16 may receive an RTCP message in block 59 indicating that a

certain number of the VoIP packets are being dropped during transmission. A user response

may also be detected in block 59 requesting an increase in sound quality.

Block 59 determines if the current congestion measurements or the current user

response requires modification of the VoIP call adaptation parameters. For example, the

20   number of lost packets identified in the congestion measurements may indicate that the sound

quality requested from the current user response is not currently being met. Alternatively, a

new user sound quality request may not be provided by the currently used adaptation

parameters.

If it is determined that adaptation parameters need to be modified, the call adaptation

25   system selects one or more adaptation parameters in block 65. The selected adaptation

parameters are compared to the congestion measurements and user response in block 67. If

the selected adaptation parameters do not provide the best chance of obtaining the sound

quality for the current user response with the current congestion conditions, the adaptation

parameters are reselected in block 65.

For example, a voice coder may be selected in block 65 that increase the number of

packets generated each second. However, there may currently be a large packet loss rate in

the VoIP call. The call adaptation system may then opt to add Forward Error Correction

(FEC) to the VoIP call instead of increasing the bit rate of the voice coder.

The selected adaptation parameters are modified according to the monitored user

response in block 69. The call adaptation system then continues to monitor congestion in

block 61 and the user response in block 63. If adaptation parameters still require

modification, the system repeats the process described above. For example, FEC may not

reduce the packet loss identified in the congestion measurements, the call adaptation system

16 may then opt to use a higher quality coder and/or vary packet payload size.

The call adaptation system uses linear regression based on current congestion

measurements to find the optimal adaptation points for each of the adaptation parameters.

The congestion measurements ensure that an adaptation envelope stays within acceptable

congestion bounds. Such bounds are usually established through either the absolute packet

loss rate or a first derivative of the absolute packet loss rate.

For example, if the packet loss rate is increasing, the call adaptation system adapts by

cutting down on total consumed bandwidth. This linear regression is conducted in block 59

of FIG. 4 where the adaptation parameters are modified according to the monitored

congestion measurements.

FIG. 5 shows another embodiment how Dual Tone Multiple Frequency (DTMF)

signals are used to detect the user response to the VoIP call. A phone 72 is connected to a

standard Plain Old Telephone Service (POTS) line 73. A user of phone 72 calls up a gateway

5   78 through a Central Office (CO) switch 76. The gateway 78 establishes a VoIP call 88 over

a packet switched network 86 to another phone 90. The gateway 78 converts analog voice

signals from phone 72 into VoIP packets.

The user of telephone 72 may wish to modify the sound quality of the VoIP call 88 as

described above. The user generates a hook flash in phone 72 to first indicate a mode change

10   to call adaptation controller 84. The user of phone 72 then presses buttons on the phone 72 to

generate the DTMF signals 74. The pressed buttons identify the user response. For example,

the "1" button by indicate a first sound quality request, the "2" button may indicate a higher

sound quality request, etc. The CO switch 76 forwards the DTMF signals 74 to gateway 78.

The call adaptation controller 84 detects the DTMF signals 84 and modifies the adaptation

15   parameters 82 in an attempt to conform the quality of the VoIP call 88 to the user response

represented by the DTMF signals 74. A Digital Signal Processor (DSP) 80 then uses the

adaptation parameters 82 selected by call adaptation controller 84 to encode and transmit

packets in VoIP call 88.

20   Graphical User Interface

FIG. 6 shows a computer 91 that includes a computer screen 92 and a keyboard 104.

The computer 91 includes an audio communication device, such as two-way speaker 106 that

allows a user to conduct a VoIP call. The user initiates the call by either dialing up a phone

number or entering an Internet address of another VoIP endpoint. After the call is

25   established, the user communicates using the two-way speaker 106, a headset, or alternative

communication device.

During the VoIP call, the user can request the call adaptation controller 108 in

computer 91 to vary sound quality of the VoIP call. A slider 96 is displayed on the screen 92

5   and allows the user to request changes in the sound quality. The user, either through a

mouse 102 or the keyboard 104, moves a pointer 100 on slider 96. When pointer 100 is

moved all the way to the left, the call adaptation controller 108 is notified that the user wants

the highest sound quality. The user moves pointer 100 all the way to the right to request the

lowest sound quality for the VoIP call.

10   In an Internet Service Provider environment, the service provider can offer different

billing regimes for differing VoIP quality. The user can use inputs to the call adaptation

controller 108 to see or control how much is being charged either before or during the VoIP

call.

          For example, the computer 91 also displays a slider 94 on screen 92 that identifies the

15   cost of the VoIP call. The cost of the VoIP call may vary according to the used bandwidth,

time of day, location of the VoIP endpoints, reservation of network resources, assigned call

priority, etc. In one example, the cost is displayed on a cost per minute basis.

          The user sees real time the result of requesting higher sound quality by viewing the

cost slider 94. If the user is satisfied with the current sound quality, but would like to reduce

20   call cost, the user can move pointer 100 to the right. If sound quality is inferior and the user

does not object to increasing the call cost, the pointer 100 can be moved to the left.

          The user can also move pointer 98 on slider 94 to limit the cost of the VoIP call. The

pointer 98 is moved to the maximum amount that the user is willing to pay for the VoIP call.

The call adaptation controller 108 then modifies the adaptation parameters to try and get the

25   best possible sound quality for the selected call cost.

          The devices used in the system described above can use dedicated processor systems,

micro controllers, programmable logic devices, or microprocessors that perform some or all

of the adaptation and voice processing operations. Some of the operations described above may be implemented in software and other operations may be implemented in hardware.

For the sake of convenience, the operations are described as various interconnected functional blocks or distinct software modules. This is not necessary, however, and there may be cases where these functional blocks or modules are equivalently aggregated into a single logic device, program or operation with unclear boundaries. In any event, the functional blocks and software modules or features of the flexible interface can be implemented by themselves, or in combination with other operations in either hardware or software.

5   Having described and illustrated the principles of the invention in a preferred embodiment thereof, it should be apparent that the invention may be modified in arrangement and detail without departing from such principles. I claim all modifications and variation coming within the spirit and scope of the following claims.